

**Listing of the Claims:**

The following is a complete listing of all the claims in the application, with an indication of the status of each:

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- 1        1 (Previously Amended). A speech coding apparatus including at least:  
2                a spectrum parameter calculation section for receiving a speech  
3        signal, obtaining a spectrum parameter, and quantizing the spectrum  
4        parameter,  
5                an adaptive codebook section for obtaining a delay and a gain from  
6        a past quantized sound source signal by using an adaptive codebook, and  
7        obtaining a residue by predicting a speech signal, and  
8                a sound source quantization section for quantizing a sound source  
9        signal of the speech signal by using the spectrum parameter and outputting  
10       the sound source signal, comprising:  
11               a discrimination section for discriminating a voiced sound mode  
12       and an unvoiced sound mode on a basis of a past quantized gain of an  
13       adaptive codebook;  
14               a sound source quantization section which has a codebook for  
15       representing a sound source signal by a combination of a plurality of non-  
16       zero pulses and collectively quantizing amplitudes or polarities of the  
17       pulses based on an output from said discrimination section, and searches  
18       combinations of code vectors stored in said codebook and a plurality of  
19       shift amounts used to shift positions of the pulses so as to output a  
20       combination of a code vector and shift amount which minimizes distortion  
21       relative to input speech; and  
22               a multiplexer section for outputting a combination of an output  
23       from said spectrum parameter calculation section, an output from said  
24       adaptive codebook section, and an output from said sound source  
25       quantization section.

1        2 (Previously Amended). A speech coding apparatus including at least:  
2                a spectrum parameter calculation section for receiving a speech  
3        signal, obtaining a spectrum parameter,  
4                an adaptive codebook section for obtaining a delay and a gain from  
5        a past quantized sound source signal by using an adaptive codebook, and  
6        obtaining a residue by predicting a speech signal, and  
7                a sound source quantization section for quantizing a sound source  
8        signal of the speech signal by using the spectrum parameter and outputting  
9        the sound source signal, comprising:  
10               a discrimination section for discriminating a voice soundmode and  
11        an unvoiced sound mode on a basis of a past quantized gain of an adaptive  
12        codebook;  
13               a sound source quantization section which has a codebook for  
14        representing a sound source signal by a combination of a plurality of non-  
15        zero pulses and collectively quantizing amplitudes or polarities of the  
16        pulses based on an output from said discrimination section, and outputs a  
17        code vector that minimizes distortion relative to input speech by generating  
18        positions of the pulses according to a predetermined rule; and  
19               a multiplexer section for outputting a combination of an output  
20        from said spectrum parameter calculation section, an output from said  
21        adaptive codebook section, and an output from said sound source  
22        quantization section.

1        3 (Previously Amended). A speech coding apparatus including at least:  
2                a spectrum parameter calculation section for receiving a speech  
3        signal, obtaining a spectrum parameter, and quantizing the spectrum  
4        parameter,  
5                an adaptive codebook section for obtaining a delay and a gain from a  
6        past quantized sound source signal by using an adaptive codebook, and  
7        obtaining a residue by predicting a speech signal, and

8 a sound source quantization section for quantizing a sound source  
9 signal of the speech signal by using the spectrum parameter and outputting  
10 the sound source signal, comprising:

11 a discrimination section for discriminating a voice sound mode and  
12 an unvoiced sound mode on the basis of a past quantized gain of an  
13 adaptive codebook;

14 a sound source quantization section which has a codebook for  
15 representing a sound source signal by a combination of a plurality of non-  
16 zero pulses and collectively quantizing amplitudes or polarities of the  
17 pulses based an output from said discrimination section, and a gain  
18 codebook for quantizing gains, and searches combinations of code vectors  
19 stored in said codebook, a plurality of shift amounts used to shift positions  
20 of the pulses, and gain code vectors stored in said gain codebook so as to  
21 output a combination of a code vector, shift amount, and gain code vector  
22 which minimizes distortion relative to input speech; and

23 a multiplexer section for outputting a combination of an output  
24 from said spectrum parameter calculation section, an output from said  
25 adaptive codebook section, and an output from said sound source  
26 quantization section.

1 4 (Previously Amended). A speech coding apparatus including at least:

2 a spectrum parameter calculation section for receiving a speech  
3 signal, obtaining a spectrum parameter, and quantizing the spectrum  
4 parameter,

5 an adaptive codebook section for obtaining a delay an a gain from a  
6 past quantized sound source signal by using an adaptive codebook, and  
7 obtaining a residue by predicting a speech signal, and

8 a sound source quantization section for quantizing a sound source  
9 signal of the speech signal by using the spectrum parameter and outputting  
10 the sound source signal, comprising:

11 a discrimination section for discriminating a voice sound mode and  
12 an unvoiced sound mode on the basis of a past quantized gain of an  
13 adaptive codebook;

14 a sound source quantization section which has a codebook for  
15 representing a sound source signal by a combination of a plurality of non-  
16 zero pulses and collectively quantizing amplitudes or polarities of the  
17 pulses based on an output from said discrimination section indicates a  
18 predetermined mode, and a gain codebook for quantizing gains, and  
19 outputs a combination of a code vector and gain code vector which  
20 minimizes distortion relative to input speech by generating positions of the  
21 pulses according to a predetermined rule; and

22 a multiplexer section for outputting a combination of an output  
23 from said spectrum parameter calculation section, an output from said  
24 adaptive codebook section, and an output from said sound source  
25 quantization section.

5 (Canceled).

1 6 (Previously Amended). A speech coding/decoding apparatus comprising:

2 a speech coding apparatus including:

3 a spectrum parameter calculation section for receiving a speech  
4 signal, obtaining a spectrum parameter, and quantizing the spectrum  
5 parameter,

6 an adaptive codebook section for obtaining a delay and a gain from  
7 a past quantized sound source signal by using an adaptive codebook, and  
8 obtaining a residue by predicting a speech signal,

9 a sound source quantization section for quantizing a sound source  
10 signal of the speech signal by using the spectrum parameter and outputting  
11 the sound source signal,

12 a discrimination section for discriminating a voice sound mode and

13 an unvoiced sound mode on the basis of a past quantized gain of a adaptive  
14 codebook, and

15 a codebook for representing a sound source signal by a  
16 combination of a plurality of non-zero pulses and collectively quantizing  
17 amplitudes or polarities of the pulses when an output from said  
18 discrimination section indicates a predetermined mode,

19 said sound source quantization section searching combinations of  
20 code vectors stored in said codebook and a plurality of shift amounts used  
21 to shift positions of the pulses so as to output a combination of a code  
22 vector and shift amount which minimizes distortion relative to input  
23 speech, and further including

24 a multiplexer section for outputting a combination of an output  
25 from said spectrum parameter calculation section, an output from said  
26 adaptive codebook section, and an output from said sound source  
27 quantization section; and

28 a speech decoding apparatus including at least:

29 a demultiplexer section for receiving and demultiplexing a  
30 spectrum parameter, a delay of an adaptive codebook, a quantized gain,  
31 and quantized sound source information,

32 a mode discrimination section for discriminating a mode by using a  
33 past quantized gain in said adaptive codebook,

34 a sound source signal reconstructing section for reconstructing a  
35 sound source signal by generating non-zero pulses from the quantized  
36 sound source information when an output from said discrimination  
37 indicates a predetermined mode, and

38 a synthesis filter section which is constituted by spectrum  
39 parameters and reproduces a speech signal by filtering the sound source  
40 signal.

1 7 (Previously Amended). A speech coding/decoding apparatus comprising:

2 a speech coding apparatus including:  
3 a spectrum parameter calculation section for receiving a speech  
4 signal, obtaining a spectrum parameter, and quantizing the spectrum  
5 parameter,  
6 an adaptive codebook section for obtaining a delay and a gain from  
7 a past quantized sound source signal by using an adaptive codebook, and  
8 obtaining a residue by predicting a speech signal,  
9 a sound source quantization section for quantizing a sound source  
10 signal of the speech signal by using the spectrum parameter and outputting  
11 the sound source signal,  
12 a discrimination section for discriminating a voice sound mode and  
13 an unvoiced sound mode on the basis of a past quantized gain of an  
14 adaptive codebook, and  
15 a codebook for representing a sound source signal by a  
16 combination of a plurality of non-zero pulses and collectively quantizing  
17 amplitudes or polarities of the pulses based on an output from said  
18 discrimination section,  
19 said sound source quantization section outputting a combination of  
20 a code vector and shift amount which minimizes distortion relative to input  
21 speech by generating positions of the pulses according to a predetermined  
22 rule, and further including  
23 a multiplexer section for outputting a combination of an output  
24 from said spectrum parameter calculation section, an output from said  
25 adaptive codebook section, and an output from said sound source  
26 quantization section; and  
27 a speech decoding apparatus including at least:  
28 a demultiplexer section for receiving and demultiplexing a  
29 spectrum parameter, a delay of an adaptive codebook, a quantized gain,  
30 and quantized sound source information,  
31 a mode discrimination section for discriminating a mode by using a

32 past quantized gain in said adaptive codebook,  
33 a sound source signal reconstructing section for reconstructing a  
34 sound source signal by generating positions of pulses according to a  
35 predetermined rule and generating amplitudes or polarities for the pulses  
36 from a code vector when an output from said discrimination section  
37 indicates a predetermined mode, and  
38 a synthesis filter section which includes spectrum parameters and  
39 reproduces a speech signal by filtering the sound source signal.

1 8 (Previously Amended). A speech coding apparatus comprising:  
2 a spectrum parameter calculation section for receiving a speech  
3 signal, obtaining a spectrum parameter, and quantizing the spectrum  
4 parameter;  
5 means for obtaining a delay and a gain from a past quantized sound  
6 source signal by using an adaptive codebook, and obtaining a residue by  
7 predicting a speech signal; and  
8 mode discrimination means for receiving a past quantized adaptive  
9 codebook gain and performing mode discrimination associated with a  
10 voiced/unvoiced mode by comparing the gain with a predetermined  
11 threshold, and  
12 further comprising:  
13 sound source quantization means for quantizing a sound source  
14 signal of the speech signal by using the spectrum parameter and outputting  
15 the signal, and searching combinations of code vectors stored in a  
16 codebook for collectively quantizing amplitudes or polarities of a plurality  
17 of pulses in a predetermined mode and a plurality of shift amounts used to  
18 temporally shift a predetermined pulse position so as to select a  
19 combination of an index of a code vector and a shift amount which  
20 minimizes distortion relative to input speech;  
21 gain quantization means for quantizing a gain by using a gain

22 codebook; and  
23 multiplex means for outputting a combination of outputs from said  
24 spectrum parameter calculation means, said adaptive codebook means, said  
25 sound source quantization means, and said gain quantization means.

1 9 (Original). An apparatus according to claim 8, wherein said sound source  
2 quantization means uses a position generated according to a predetermined  
3 rule as a pulse position when mode discrimination indicates a  
4 predetermined mode.

1 10 (Original). An apparatus according to claim 9, wherein when mode  
2 discrimination indicates a predetermined mode, a predetermined number of  
3 pulse positions are generated by random number generating means and  
4 output to said sound source quantization means.

1 11 (Original). An apparatus according to claim 8, wherein when mode  
2 discrimination indicates a predetermined mode, said sound source  
3 quantization means selects a plurality of combinations from combinations  
4 of all code vectors in said codebook and shift amounts for pulse positions  
5 in an order in which a predetermined distortion amount is minimized, and  
6 outputs the combinations to said gain quantization means, and  
7 said gain quantization means quantized a plurality of sets of  
8 outputs from said sound source quantization means by using said gain  
9 codebook, and selects a combination of a shift amount, sound source code  
10 vector, and gain code vector which minimizes the predetermined distortion  
11 amount.

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